SIP

- 2PORTS
- 4PORTS
- 8PORTS

OPERATION MANUAL

Operation Manual

Contents

1. Introduction

Product Overview

The stand-alone VoIP Gateway carries both voice and facsimile over the IP network. It supports SIP industry standard call control protocol to be compatible with free registration services or VoIP service providers' systems. It works in two different modes: UA (User Agent) or Server. As a standard user agent, it is compatible to all well-known Soft Switches and SIP proxy servers. While running the optional server software, the gateway can be configured to establish a private VoIP network over the Internet without a 3rd party SIP Proxy Server.

There are 3 types of gateways in the same series: 2 ports, 4 ports and 8 ports (voice ports, FXS and/or FXO). The gateway can be seamlessly integrated to existing network by connecting to a phone set, PBX, key telephone system, fax machine or PSTN line. With only a broadband connection such as ADSL bridge/router, Cable Modem or leased line router, it allows you to gain access to voice and fax services over the IP in order to reduce the cost of international and long distance calls.

In addition, the in-built 4 ports Ethernet switch supports comprehensive Internet gateway functions to accommodate other PCs or IP devices to share the same broadband stream. QoS function allows voice and data traffic to flow through where voice traffic is transmitted in the highest priority. With TOS bit enabled, it guarantees voice packets to have first priority to pass through a TOS enabled router.

With the support of DDNS, it makes the gateway reachable by its domain name where the ISP dynamically assigns the IP address. It helps users to host a web site or mail server in a PPPoE or DHCP network. By enabling the CDR function & setting up a simple server, administrators are allowed to log in and view all call records such as call duration, time and date of calls and latency.

The gateway can be assigned with a fixed IP address or by DHCP, PPPoE. It adopts the G.711, G.726, G.729A or G.723.1 voice compression format to save the network bandwidth while providing real-time and toll quality voice. In addition, in the event that the power supply fails or Internet connection is lost, the gateway can automatically divert the FXS end to the PSTN network on the FXO port so users can still use the conventional PSTN line to make calls. This feature is especially useful while dialing emergency calls (i.e. 911).

Product Features

- SIP (RFC 3261) compliant
- Supports 2 / 4 / 8 simultaneous FAX / Voice calls (port number differs between models)
- 4 Ethernet switch ports with IP sharing functions
- Optional server enables small businesses to build up private VoIP network (SIP model)
- QoS support guarantees voice bandwidth in a busy network
- Supports IP TOS (Type Of Service)
- Internet gateway functions: DHCP server, NAT, Virtual Server, DMZ, IP/PORT/MAC filtering
- T.30 (G III) / Real Time T.38 / Secured T.38 Fax Relay
- Feasible for Fixed IP address or dynamic IP address network (PPPoE / DHCP client, support DDNS)
- Configurable Hot Line feature
- Supports IP-to-PSTN / PSTN-to-IP applications
- Life-Line support (IP / power failure over relay)
- NAT traversal - STUN and UPnP (optional)
- Pass through NAT
- Call Detailed Records (CDR)
- Web-based firmware upgrade
- Caller ID Delivery
- Easy Configuration by IVR and Web-based GUI
- Greeting message
- Echo Cancellation: G.168 compliant
- Voice Activity Detection (VAD) & Comfort Noise Generation (CNG)
- Dialing: DTMF, PULSE (optional)
- Signaling Protocol: Loop Start
- Adaptive Jitter Buffer and Programmable Gain Control
- Internet Gateway Functions : DHCP server, NAT, Virtual Server, DMZ, IP/PORT/MAC filtering

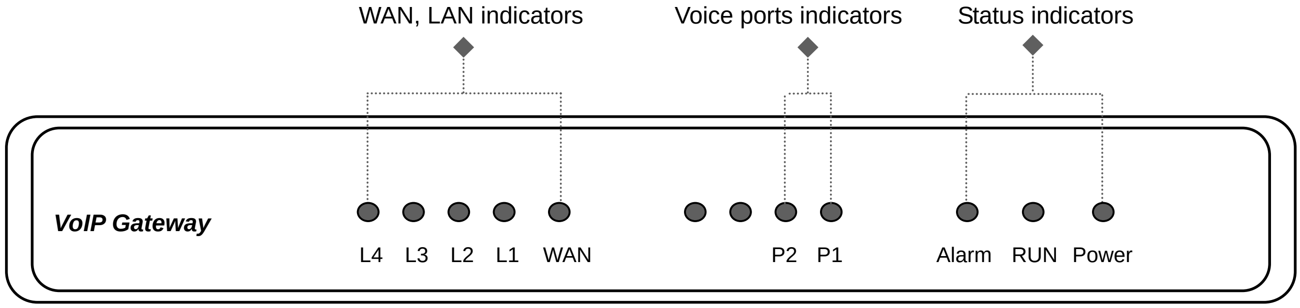
CALL features (optional)

- Call hold
- Call waiting
- Call forward
  - Unconditional (follow me)
  - Busy forward
  - No answer forward
- Call transfer

Hardware Description

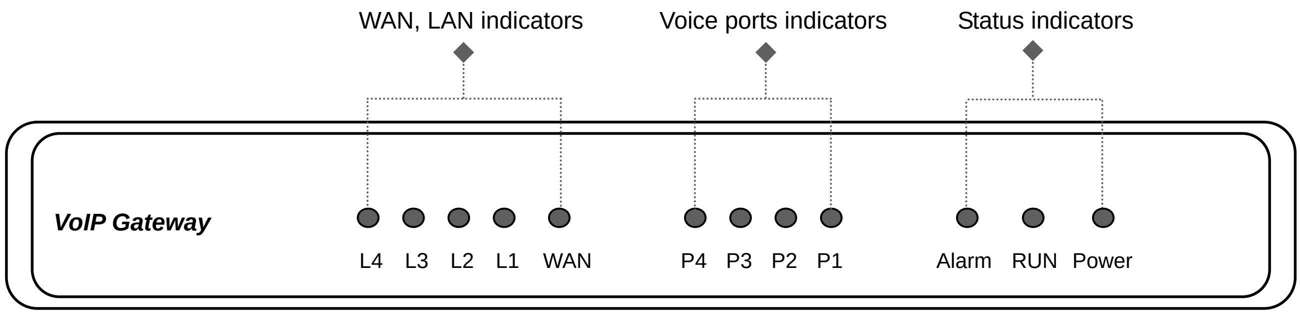
2 ports gateway model : 1S1O

Front Panel



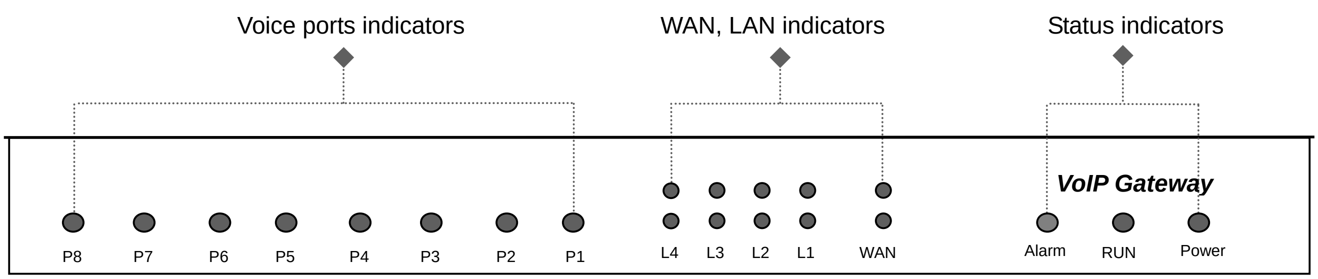
4 ports gateway model : 2S2O

Front Panel



8 ports gateway model : 4S4O

Front Panel



- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- Voice ports indicators indicates the port is using.
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

- ✓ **When starting up the system, the Alarm, Run, and Power indicators will light up. After about 15 seconds, the Alarm indicator will go off, the Run indicator will blink in green, and the Power indicator will stay green under operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with ISP and has yet to obtain an IP address.**

- ✓ **When the ethereal cable is connected, the Link indicator will light up and if data is being transmitted over the Internet, the act indicator will blink.**

Restore to factory default: (IP address, User's. Name and Password)

(1) Pull off the power plug.
(2) Press reset (do not let go of the reset button).
(3) Plug the plug back into the socket (do not let go of the reset button).
(4) Let go of the reset button after 6 seconds. Factory settings will be restored.

NOTE: Do not connect FXS ports to each other. Also, do not connect any FXS port directly to a PSTN line or internal PBX. If any of these are done, your 4S may be damaged.

Please use the power adapter comes with VoIP Gateway. If using adapters other than the one comes with VoIP Gateway, it may cause problems and will affect the warranty of the product.

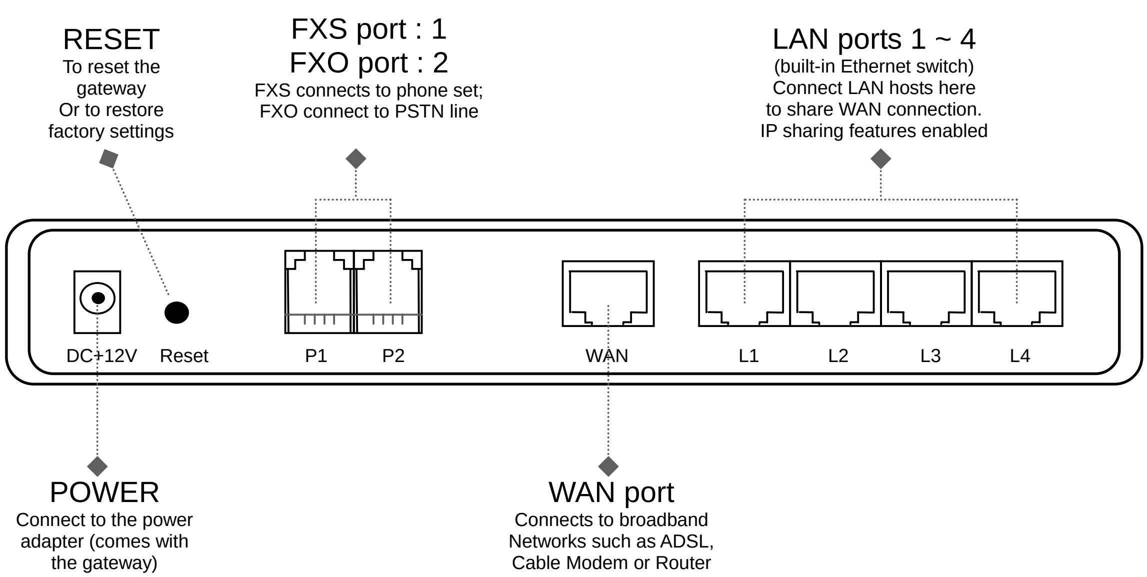
1S1O **:** P1 is a telephone port (FXS), and P2 is a line port (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P2 for emergency calls.

2S2O **:** P1-P2 is telephone ports (FXS), and P3-P4 are line ports (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P3, and P2 to P4 to reach PSTN.

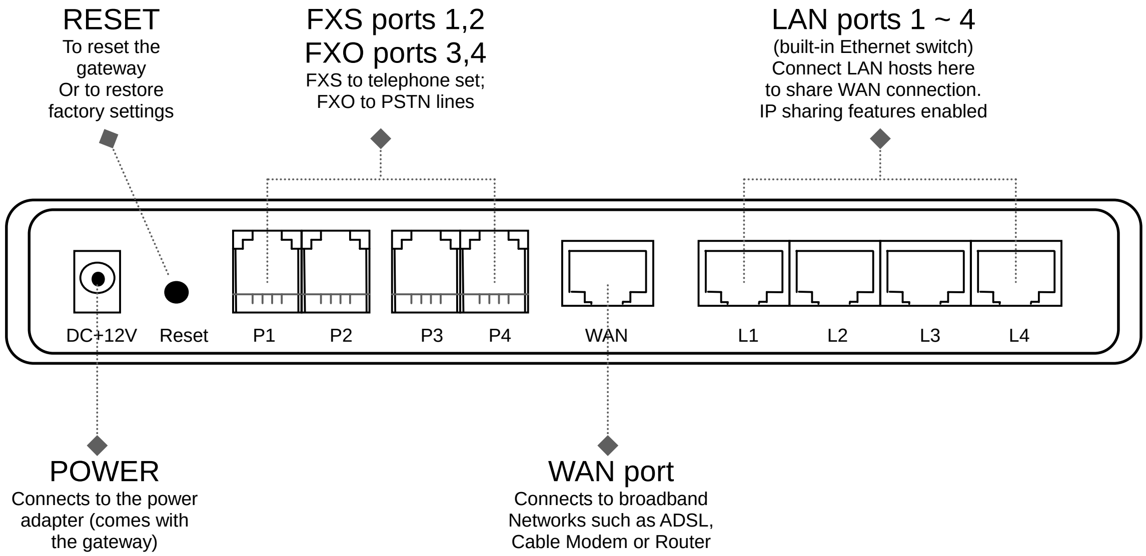
4S4O **:** P1-P4 are telephone ports (FXS), and P5-P8 are line ports (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P5, P2 relayed to P6, P3 relayed to P7, and P4 relayed to P8 to reach PSTN.

Rear Panel

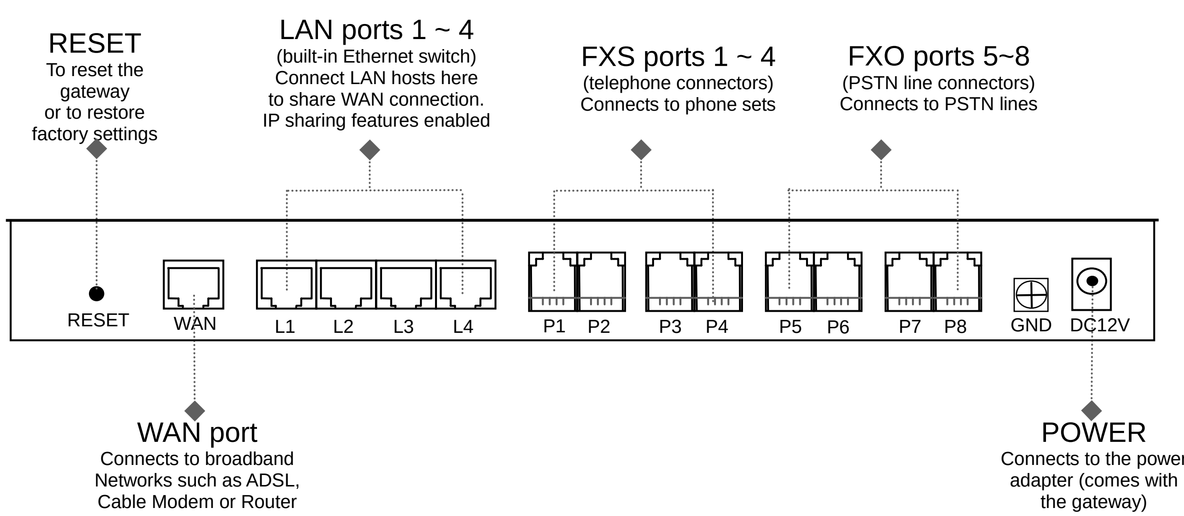
1S1O Model (1 FXO and 1 FXS ports)



2S2O Model (2 FXS and 2 FXO ports)



4S4O Model (4 FXS and 4 FXO ports)



2. Installation and Applications

Network Interface

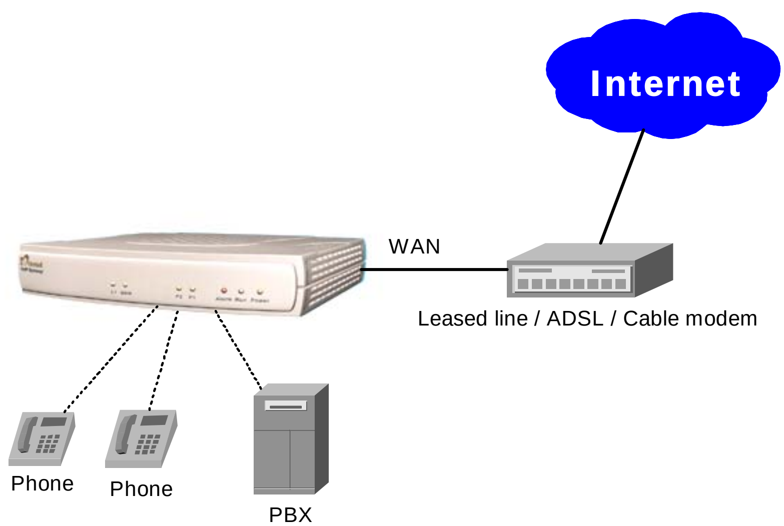
The network interface is divided into 3 basic modes as described below:

- Gateway can be assigned with a Public IP Address
- Gateway can be built under the existing NAT
- Gateway can be assigned with a Public IP address and serves as an IP sharing router.

Gateway Assigned with a Public IP Address

The gateway will have a Public IP address for Internet connection regardless of whether it is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dialup / ADSL).

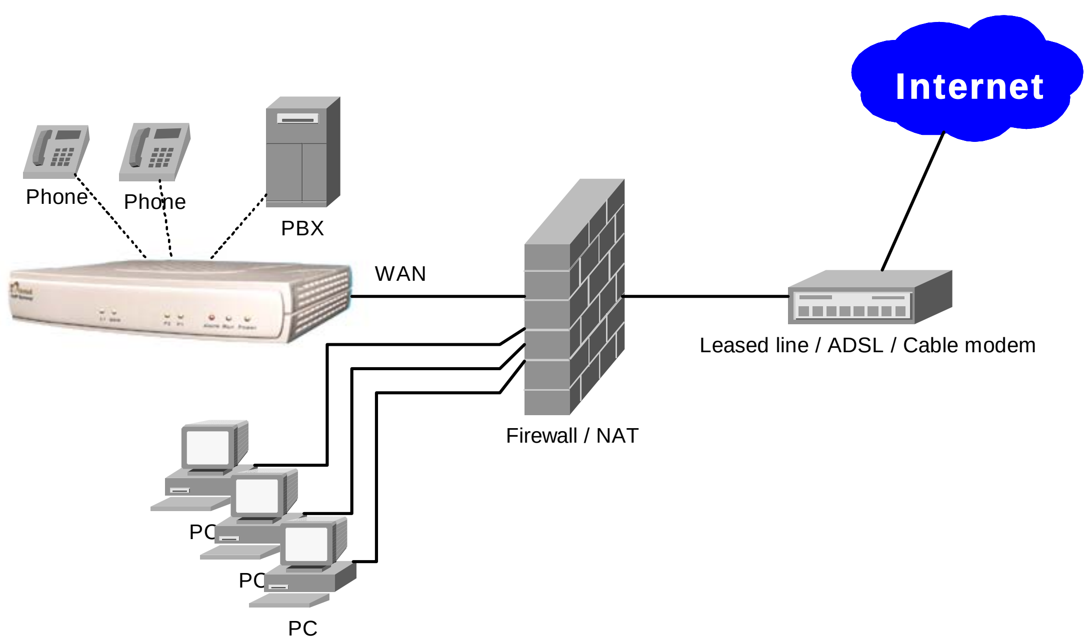
Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	



Gateway in a NAT network

The gateway uses a virtual IP address and the IP sharing function of other systems to connect to the Internet.

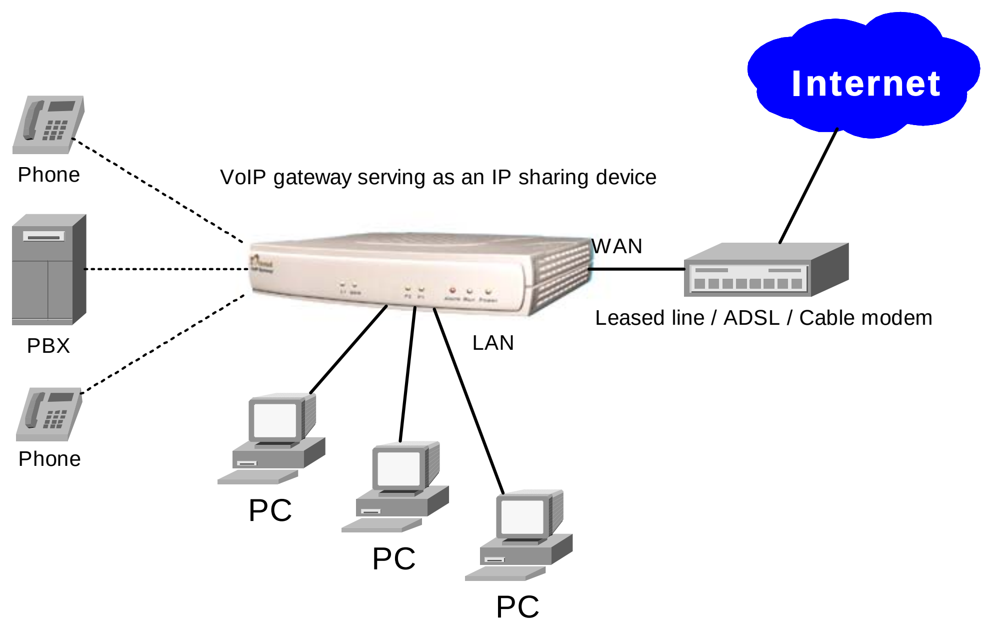
<table>
<tr><td>LAN IP address of IP sharing</td><td colspan="3">Please avoid IP address 192.168.0.1-192.168.8.254 (You may need to change the settings of IP sharing or change SIP series Gateway LAN Port IP address)</td></tr>
<tr><td>Gateway IP Settings</td><td colspan="3">Set as static IP address, and assign the LAN IP address of the IP sharing to the Default Gateway.</td></tr>
<tr><td rowspan="2">NAT /STUN Settings
Please refer to P. 29 for the Use NAT.</td><td rowspan="2">Enable</td><td colspan="2">If the WAN of the IP sharing device has static IP address, then the NAT IP address is set as the Public IP address of the IP sharing.</td></tr>
<tr><td colspan="2">If the WAN of the IP sharing device uses a dynamic IP address, then it has to comply with the DDNS settings. When suing NAT, you must enter the URL (Uniform Resource Locator) that is registered to the DDNS server.</td></tr>
<tr><td rowspan="2">DDNS Settings
Please refer to P. 29 for the DDNS Settings</td><td colspan="2">The WAN of the IP sharing device has a static IP address.</td><td>Disabled</td></tr>
<tr><td colspan="2">The WAN of the IP sharing device has a dynamic IP address.</td><td>Enabled: enter the registered URL (Uniform Resource Locator) into the network settings -> under NAT</td></tr>
</table>



Gateway assigned with a Public IP Address and serving as an IP sharing device

The gateway will have a Public IP address regardless of whether it is a static IP application, DHCP (using a Cable Modem), or PPPoE (To connect to your ADSL account), which can then use the functions of built-in IP sharing function to allow other PCs to be on-line at the same time.

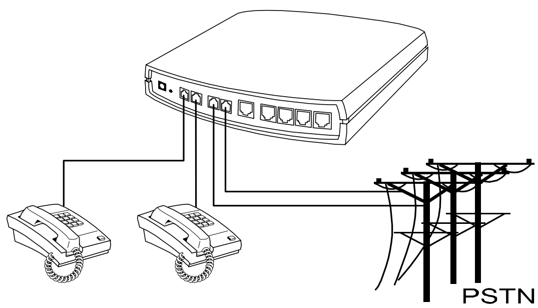
Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	
For settings at PC end, please refer to section "IP sharing functions"	PC uses a static IP address ranging from: 192.168.8.1-192.168.8.253 Subnet Mask：255.255.255.0 Default Gateway：192.168.8.254	



Telephone Interface Description (4 ports models for example)

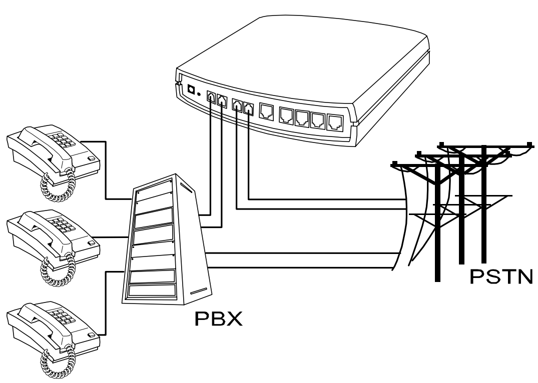
Example for 2S2O gateway:

P1-P2 is FXS interfaces and can be directly connected to a telephone set for direct calls. P3-P4 is FXO interfaces and can be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call made from a traditional telephone line to connect with a Gateway user.



Integrating the 2S2O with PBX

P1-P2 is FXS interfaces and can be connected to a PBX CO; P3-P4 is FXO interfaces and can be connected to a PSTN to act as a bridge between the PSTN and other VoIP telephones. The system also allows a call made from a traditional telephone line to connect with a Gateway user.



3. Setting the Gateway through IVR

VoIP transmits voice data (packet) via the Internet to achieve telecommunications. This means that the telecommunication quality is closely related to the whole network environment. If any one of the telecommunicating parties has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, an excellent telecommunication can only be created when Gateway is connected to the Internet and when network environment is stable.

Preparation

- Install the Gateway according to instructions. Connect the power supply, telephone set, telephone cable, and network cable properly as described in Chapter 2.
- If a static IP is used, confirm the desired IP settings of the WAN Port (IP address, Subnet Mask, and Default gateway). Please contact your local Internet Service Provider (ISP) if you have any questions.
- If using dialup ADSL (PPPoE) for network connection, confirm the dialup account number and password.
- If users wish to build Gateway under the NAT, Gateway WAN Port IP address and LAN Port should not use the same range. This is to avoid phone failures.

Basic Settings of a Gateway

- IP Settings— Connecting Gateway to the Internet.
- Telephone Number of Gateway Phone Number (Gateway and SIP Prefix Number 5368 Gateway representative number).
- Using **Phone Book** or **Phone books manager**

<u>Please refer to Section 6 for Basic Dialing Method</u>

Gateway provides two setting modes:

1. Telephone IVR Setting Mode.
2. Browser Setting Mode.

The IVR provides basic query and setting functions, while the browser provides a full setting function.

IVR (Interactive Voice Response)

Gateway provides convenient IVR functions. Users only need to pick up a handset and enter the function code for the query and setting without using a PC.

NOTE: After finishing the settings, make sure the new settings are saved. This is so that the new settings will take effect after the system is restarted.

Instructions

- FXS Port: Connected to telephones. To enter IVR mode, enter " * * password #" after hearing the dial tone. When you hear a second dial tone, the system is in IVR mode, enter the function code. (Please refer to the Advanced Settings on page 35 for these codes)

Example: The factory default code is blank. Enter **"**#"**. You are now in IVR setting mode, enter the desired code. E.g.: if the code is 1234, then enter **"**1234#"**.

- FXO Port: to use IVR functions, dial the phone number of FXO Port using an external line. You will hear the instruction "enter value", and then enter a PIN number. The factory default code is blank. Enter "**#" as above. You are now in IVR setting mode.
- Once the first setting or query has been completed, you will hear a dial tone. Then use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

Example: enter **"**#"** (You are now in IVR mode)→ enter **101** (to query IP address) → the system responds with an IP address →you can continue with more settings or queries: enter **111** (to set IP address) →enter **192*168*1*2** (IP number).

Save Settings

After entering IVR mode, dial 509 (Save Settings). Wait for about 3 seconds and after hearing a confirmation tone "1", hang up the phone. Please reboot the Gateway to enable the new settings.

To inquire about current Gateway's WAN Port IP address

After entering IVR mode, dial 101. The system will repeat the current WAN Port IP address.
If the system does not repeat the IP address, it indicates that the Gateway is not currently connected to the Internet. Please check if the cable connection, account number, and password are correct.

Software Upgrades

IVR provides online upgrades. Once in IVR mode, enter "209" and you will hear "Enter Value". Enter your IP address followed by "#" (ie.：**61*30*25*89#**). Then you will hear a second "Enter Value". Enter the Listen Port Number followed by "#"(ie.: **6001#**). For information about upgrading, please ask your agent.

IVR Functions Table:

Function Code	Description	Example
111/101	WAN Port IP address Set/Query	Use in conjunction with function code **114**, select 1 for a Static IP function.
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2.DHCP, 3.PPPoE)	
115/105	DNS IP address Set/Query	
116/106	Phone books manager IP address Set/Query	Must use 116/106, 117/107 in conjunction with each other.
117/107	Set/Query whether or not to use Public Telephone Book (0: Disable 1:Enable)	
199/099	Set/Query whether or not this Gateway acts as the phone books manager (0: Disable 1: Enable)	
066	Querying the connection to Phone books manager	
118	Restart	
121	Setting PPPoE Account	Use in conjunction with function code **114,** select 3 for a PPPoE function
122	Setting PPPoE Password	
123	Setting NAT IP address	Must use 123 and 124 in conjunction with each other.
124	Uses NAT (0: Disable 1: Enable)	
151/141	Register to Proxy Server Set/Query (0: Disable 1: Enable)	
152/142	Proxy Server IP address Set/Query	
153/143	Proxy Server Port Set/Query	
125	Set Proxy Server account	
126	Set Proxy Server password	
154/144	Uses STUN Set/Query (0: Disable 1: Enable)	
155/145	STUN IP address Set/Query	
156/146	STUN Port Set/Query	
311/301		
312/302		
131/132		
133	Saving greeting message	
211/201	Set/Query International Prefix code	Prefix dialed before making an international call e.g. 002 and 005.
212/202	Set/Query Country Code	Setting country code, e.g. 886
213/203	Set/Query Area Prefix Code (Long-Distance Prefix Code)	Prefix dialed before making a long-distance call e.g. 0.
214/204	Set/Query Area Code	eg. “2” for Node B area.

Function Code	Description	Example
215/205	Set/Query Gateway Telephone Number (Representative Number)	
216/206	Set/Query the extension number of Line 1.	
217/207		
109	Restoring factory default IP address configuration	A static IP address for WAN Port IP：192.168.1.2 Mask：255.255.255.0 Gateway：192.168.1.254
409	Restoring factory default settings	
509	Save settings	
900	Setting IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Soft Upgrade	

IP Configuration Settings—Setting IP Configuration of WAN Port

Static IP Settings

NOTE: Complete static IP settings should include a static IP (Option 1 under114), IP address (111), Subnet Mask (112), and Default Gateway (113). Please contact your local Internet Service Provider (ISP) if you have any questions.

Function	Command
Select a Static IP	• After entering IVR mode, dial 114. • After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	• After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address, followed by "#". Example: If the IP address is 192.168.1.200, dial 192*168*1*200#.
Subnet Mask Settings	• After entering IVR mode, dial 112. After hearing "Enter value", enter your netmask, followed by "#". Example: If the mask value is 255.255.255.0, dial 255*255*255*0#.
Default Gateway Settings	• After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address, followed by "#". Example: If the Default Gateway is 192.168.1.254, dial 192*168*1*254#.
Save Settings and Restart	• To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are correct.

Dynamic IP (DHCP) Settings

After entering IVR mode, dial 114.
After hearing "Enter value", dial 2 (to select DHCP).
Saving settings –press 509 (Save Settings). Please restart the system. After the system is restarted, press 101 to check if the IP address is retained.
If the system does not repeat the IP address, it indicates Gateway has not been properly connected to the Internet. Please check the cable connection.

ADSL PPPoE Settings

NOTE: Complete PPPoE settings should include: Select PPPoE (Option 3 of 114), PPPoE account (121) and PPPoE password (122).

Please contact your local Internet Service Provider (ISP) if you have any questions.

Select a PPPoE

- After entering IVR mode, dial 114.
- After hearing "Enter value", dial 3 (to select PPPoE).

PPPoE Account Settings

- After entering IVR mode, dial 121.
- After hearing “Enter value”, enter the account number, followed by ”#”.

Example: If the account is “84943122 @ hinet.net”, please enter 08 04 09 04 03 01 02 02 71 48 49 54 45 60 72 54 45 60#.

Please note that it is necessary to enter two digits for each character/number; for example, enter “01” for “1” and “11” for “A”.

PPPoE Password Setting

- After entering IVR mode, dial 122 after hearing “enter value” followed by “#”.

Example: If the password is “3ttixike”, please enter “03 60 60 49 64 49 51 45#”.

Save Settings and Restart

To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are correct.

Recorded Voice File

- The gateway allows users to record their incoming call greeting messages, when calling via FXO.
- After entering IVR mode, dial 132. After hearing “Enter value”, record the incoming call greeting message. To end recording, simply hang up.
- After recording, to listen to the recorded message, press 131. Press 133 to save the message.

PPPoE Character Conversion Table

Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	Input Key
0	00	A	11	a	41	@	71
1	01	B	12	b	42	•	72
2	02	C	13	c	43	!	73
3	03	D	14	d	44	"	74
4	04	E	15	e	45	$	75
5	05	F	16	f	46	%	76
6	06	G	17	g	47	&	77
7	07	H	18	h	48	'	78
8	08	I	19	i	49	(	79
9	09	J	20	j	50	)	80
		K	21	k	51	+	81
		L	22	l	52	,	82
		M	23	m	53	-	83
		N	24	n	54	/	84
		O	25	o	55	:	85
		P	26	p	56	;	86
		Q	27	q	57	<	87
		R	28	r	58	=	88
		S	29	s	59	>	89
		T	30	t	60	?	90
		U	31	u	61	[	91
		V	32	v	62	\	92
		W	33	w	63	]	93
		X	34	x	64	^	94
		Y	35	y	65	_	95
		Z	36	z	66	{	96
						|	97
						}	98

4. Setting a Gateway with WEB Browser

The gateway allows users to make settings using a web browser. After opening a browser, enter Gateway's IP address as the website address in order to enter the Web configuration screen as shown in the following diagram. (IE Browser used for example: Enter http：//192.168.1.2.)

The factory default WAN IP address for Gateway is **192. 168. 1. 2**.You can also enter "101" from the handset to inquire about the current WAN Port IP address. The factory default LAN Port IP address is 192.168.8.254.

Instructions

- Open an Internet browser.
- Enter gateway's WAN Port IP address in the website address area (If the PC is connected to the LAN Port, enter the LAN Port IP address. The default is 192.168.8.254)
- The following registration screen will appear (The factory default settings for **Login ID and Password are set to blank**).
- After completing and confirming the settings, some of the settings will take effect immediately. But network related settings would take effect after the gateway is restarted. Please go to **System Operation** to save the settings before restarting the system.

To avoid several people simultaneously configuring the web and causing problems to users, please enter the correct Login ID and password. If a user logs into the system, other users from different IP addresses cannot login at the same time. Please remember to logout or restart the system if not using the web configuration function.

Network Settings

The network settings are used to set the gateway's communication ports, IP configurations, and Phone Books Manager IP etc.

Network Settings (WAN)

Current WAN IP Address	192.168.1.3		
Listen Port UDP [1 - 65535]	5060	RTP Starting Port UDP [1 - 65500]	9000

DHCP ○		
Static IP ◉	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254
PPPoE ○	PPPoE Account	
	PPPoE Password	
	Confirm Password	
PPTP ○	IP address	
	Subnet mask	
	PPTP Server	
	PPTP ID	
	PPTP Password	
	Confirm Password	
BigPond Cable ○	User Name	
	BigPond Cable Password	
	Confirm Password	
	Login Server	
Domain Name Server Assignment	○ Auto ◉ Manual	
Domain Name Server (Primary) IP	192.168.1.254	Domain Name Server (Secondary) IP:

- Current WAN IP Address: The IP address of WAN port.
- Listen Port UDP: It is not necessary to change the protocol of the communication port used by VoIP Gateway.

- RTP Starting Port UDP: The initial value of port number for transmitting voice data among Gateway(s). Each line requires 2 ports. It is not necessary to change these.

For example, If the starting port is 9000, then Line 1 is using 9000 and 9001, and Line 2 is using 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

There are four methods of obtaining a WAN port IP address:

1. Static IP
2. DHCP, means a Dynamic IP (Cable Modem)
3. PPPoE (Dialup ADSL)
4. PPTP.

Using the DHCP and PPPoE for obtaining an IP address may vary. If not familiar with the network connection, please contact your local ISP.

Setting Dynamic IP (DHCP)

DHCP ⊙	

Click “DHCP” to obtain a Dynamic IP address, then click the “Accept” button at the bottom of the screen.
Save the settings: Click **System Operation** to select “Save Settings”,
”Restart”, and then click the ”Accept” button. Wait for a while (about 40 seconds), and the system will obtain the related IP value from the DHCP Server.

NOTE: After the system has obtained a new IP address, if using WAN Port to enter the Web Configuration Screen, a new IP address has to be used. The same applies to the following two settings.

Setting Static IP

Static IP ⊙	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254

Select “Static IP” and enter the IP address, Subnet Mask and Default Gateway values. Then click the “Accept” button at the bottom of the screen.
Save the settings, and then restart the system. Wait for about 40 seconds for the system to restart.

ADSL PPPoE Settings

PPPoE ⊙	PPPoE Account	PPPoE
	PPPoE Password	****
	Confirm Password	****

Select “PPPoE” Enter the Account Number, Password and Reenter Password to confirm. Then click the “Accept” button at the bottom.
Save the settings, and then restart the system. The system will take about 49 seconds to restart.

PPTP※

PPTP ⦿	IP address	66.77.88.99
	Subnet mask	255.255.255.0
	PPTP Server	66.77.88.1
	PPTP ID	PPTP ID
	PPTP Password	**
	Confirm Password	**

Select “PPTP” and enter the IP Address, Subnet mask, PPTP Server, PPTP ID and Password. Then click the “Accept” button at the bottom.

Save the settings, and then restart the system. The system will take about 40 seconds to restart.

If not familiar with the network connection, please contact your local ISP.Domain Name Server

BigPond (for Australia use only)

BigPond Cable ○	User Name	
	BigPond Cable Password	
	Confirm Password	
	Login Server	

Click “BigPond Cable” Enter User Name and Password. Login Server is optional. Then click the “Accept” button at the bottom.

(DNS) Settings

Domain Name Server Assignment	○ Auto ⦿ Manual		
Domain Name Server (Primary) IP	60.12.34.56	Domain Name Server (Secondary) IP	60.78.90.12

Domain Name Server (DNS): While a gateway is accessing another gateway or computer with a hostname, it will look up the IP address from the DNS provided by ISP. The ISP whilst negotiating with PPPoE or DHCP usually assigns the DNS information. In the case that the DNS is not assigned automatically or WAN port is assigned with a static IP address, the DNS information must be assigned manually.

Auto : Gateway learns primary & secondary addresses from ISP’s DHCP server or PPPoE server.

Manual : Enter the primary & secondary addresses manually. Please be sure the IP addresses are correct otherwise the gateway will not be able to access hosts with its hostname.

Clone MAC

Factory Default MAC Address	000C2A050341	Restore
Your MAC Address	00001CD4AF0A	Clone
Current MAC Address	000C2A050341	

Some Internet Service Providers (ISP) assigns the bandwidth via the MAC (Media Access Control) Address. You can click the " Clone" button to copy the MAC address of the Ethernet Card installed in the computer used to configure the device. It is only necessary to fill in the field if required by your ISP.
The “Your MAC Address” will be blank as you log in through WAN port.

Using Phone Books Manager

Enable Phone Book Manager Server	☐ Clients List		
Share Local Phone Book to Clients	☐	TTL (Expire time: mins) [1 - 60]	1
Register to Phone Book Manager	☐	VoIP failure announcement	☐
Gateway Name for Phone Book Manager			
Phone Book Manager Login Password		Confirm Password	
Phone Book Manager IP/Domain	192.168.1.1	Phone Book Manager Server Listen Port [1 - 65535]	1690

- Enable Phone Books Manager Server: It allows other Gateway users to register the IP address and telephone number in this Phone books manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- Share Local Phone Book: While this option is enabled and the gateway is performing as a Phone Books Manager, this gateway will append its local Phone Book entries to the Manager for other clients to lookup.
- TTL (Time to Live): If a Gateway system that is controlled by the Phone Books Manager does not report back within the deadline set by TTL, the system will be excluded from the user’s list. Each Gateway should report to the Phone Books Manager once every 30 seconds.
- Register to Phone Books Manager: To register to the Phone Books Manager.
- VoIP failure announcement: If a Gateway fails to register to the Phone Book Manager, it will play a voice announcement when FXS is pick-up.
- Gateway Name for Phone Book Manager: The alias registered with the Phone Books Manager.
- Phone Books Manager Login Password: Enter the registered password. If this system is serving as the Phone Books Manager, the set password is also the password used for registering other Gateway systems.
- Phone Books Manager IP/Domain: Enter the IP address for the Phone Books Manager. It supports URL (Uniform Resource Locator).
- Phone Books Manager Port: The protocol communication port for transmitting signals between the Phone Books Manager and other Gateway systems. Please confirm whether the setting is the same as that of the **Phone Books Manager.**

LAN interface mode
◉ Router ○ Bridge

Network Settings (LAN)			
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0
DHCP Server			
Enable DHCP Server	☐		
IP Pool Starting Address	192.168.8.1	IP Pool Ending Address	192.168.8.250
Lease Time [1 - 9999 hours]	1		
Domain Name Server Assignment	◉ Auto ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

Port of Web Access from WAN [1 - 65535]	80

LAN interface mode※

LAN interface mode
◉ Router ○ Bridge

- Router: The system serves as a router.
- Bridge: The system serves as a bridge between WAN port and LAN port.

Network Settings (LAN)

LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0

- Network Settings (LAN): Gateway LAN Port IP address and the subnet mask value. Please note that Gateway is built under NAT: <u>Gateway LAN Port IP address cannot be in the same section as the NAT LAN Port IP address,</u> or else it is unable to make or receive calls. For example, if the NAT LAN Port IP address is 192.168.8.1, then Gateway LAN Port cannot be ranged between 192.168.8.1 ~ 192.168.8.254. If so, please change the LAN port IP address, (e.g. setting the IP address to 192.168.99.254.)

DHCP Server			
Enable DHCP Server	☑		
IP Pool Starting Address	192.168.8.1	IP Pool ending Address	192.168.8.250
Lease Time[1 - 9999hours]	1		

- Enable DHCP Server: Enable or Disable DHCP server service of gateway.
- IP Pool Starting Address: The first IP address to be assigned to DHCP clients.
- IP Pool ending Address: The last IP address to be assigned to DHCP clients.
- Lease Time: The valid period of an assigned IP address.

Domain Name Server Assignment	◉ Auto ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

- Domain Name Server Assignment: The DNS information to be assigned to DHCP clients.
  Auto : Assigns the same DNS information of WAN port to the DHCP clients.
  Manual : Manually assigns the DNS information for DHCP clients.

Port of Web Access from WAN[0 - 65535]	80

- Port of Web Access from WAN: Http port for WAN. To make this setting, the LAN Port must be used. Settings cannot be made using the WAN Port. Always use port 80 when connecting to LAN port.

QoS Settings

WAN QoS

WAN QoS		
☐ QoS	Downstream Bandwidth	Full
	Upstream Bandwidth	Full
ToS / DiffServ Settings		
ToS IP Precedence ◉	Signaling Precedence	3 (Flash)
	Voice Data Precedence	5 (CRITIC / ECP)
DiffServ (DSCP) ○	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31)
	Voice Data Value	46 (Expedited Forwarding, EF)

- QoS (Quality of Service): Sets an external bandwidth to ensure sound quality during transmission (When this function is enabled, the voice packet has the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission). Some models of the VoIP Gateway without this function can adjust the bandwidth automatically.
- ToS/DiffServ(Type of Service/DSCP): The voice packet has the highest priority to ensure telecommunication quality; the larger the value you set, the higher priority you will get.

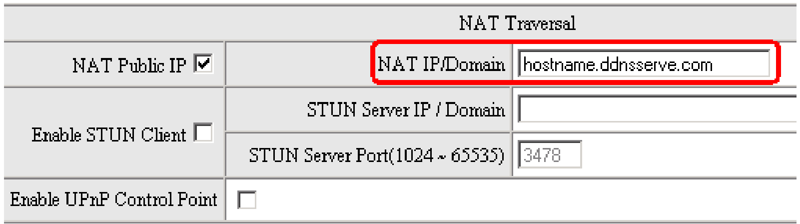
LAN QoS※

LAN QoS				
Enable LAN QoS	☐			
Port #	Priority	Flow Control	Incoming Rate Limit	Outgoing Rate Limit
LAN Port 1	Low	☐	Full	Full
LAN Port 2	Low	☐	Full	Full
LAN Port 3	Low	☐	Full	Full
LAN Port 4	Low	☐	Full	Full

- Priority: Set the priority of physic LAN port.
- Flow Control: Enable or Disable Flow control.
- Incoming Rate Limit: Set the incoming rate limit of a specific LAN port.
- Outgoing Rate Limit: Set the outgoing rate limit of a specific LAN port.

NAT/DDNS

NAT Traversal



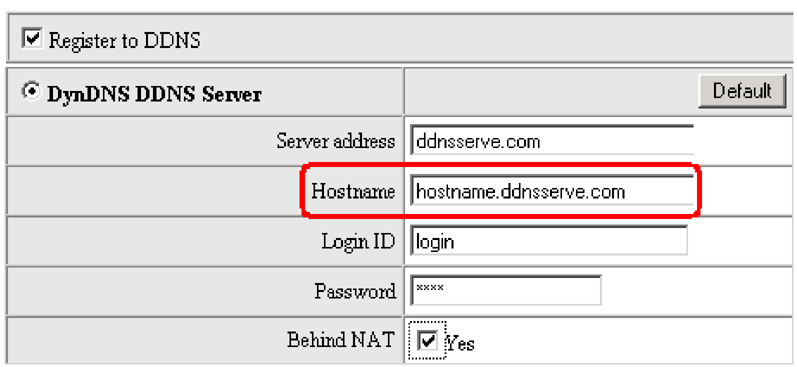
If a Gateway is set up under an IP sharing setting, you can select either the NAT or STUN protocol.

- NAT Public IP: The IP address used by the gateway should be a virtual address. Further more, users must set the Virtual Server Mapping in the NAT Server (A virtual server is defined as a Service Port, and all requests to this port will be redirected to this specified the server IP address).

  The default port is listed below:
  Listen Port (UDP): 5060
  RTP Starting Port (UDP): 9000~9015 (Listen Port used for telephone communication).
  Http Port (TCP): 80

- NAT IP/Domain: Enter the NAT Server IP address (Real External IP address of NAT Server). If using DDNS (Please refer to the next setting item), then fill in the URL (Uniform Resource Locator).
- STUN: Use STUN protocol prevents problems setting IP sharing, but some NAT do not support this protocol.
- STUN Server and Port: Enter the STUN server IP address and Listen Port number.
- Uses UPnP: Add a new function to enable the VoIP gateway's IP traffic to pass through a NAT server. This function only works when the NAT server support UPnP and has it enabled.

DDNS



These settings are only necessary when the gateway is set up under a NAT that uses a dynamic IP address and do not support DDNS.
Choose a DDNS Server: The current system allows users to choose either DynDNS、TZO、3322.org、PeanutHull or a private server. Please apply for a user account before choosing a service provider.

- Server address: Sets up the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- Hostname: The URL of the system (or NAT) – apply from a domain name registration providers.
- Login ID and Password: The ID and password are used to log into the DDNS server.
- Behind NAT: Select only when the system is set up under NAT.

NOTE: If the Gateway is set up under NAT, then enter the hostname into the NAT IP/Domain.

Telephony Settings

Prefix Number Rules

Prefix Number Rules	
Trunk Dial Out Verify	01;00
Trunk Dial Out Replace	190601;190200
Trunk Dail Out Deny	020

- Trunk Dial Out Verify/ Trunk Dial Out Replace: The system will transfer the number for all transit out call through FXO port. For example: If you transit out with 01907123456, the system will trans to 190601 907123456. If you transit out with 008621123456 the system will replace it with 190200 8621123456.
- Trunk Dial Out Deny: The system will deny the call with the leading number filled in this column.

FXS Caller ID Generation	(•) Disable ( ) DTMF ( ) FSK		
FXO Caller ID Detection	[x]	Detection Level	0
FSK Caller ID Type	(•) Bellcore ( ) ETSI		
Anonymous Caller ID (CLIR)	[ ]		
Anonymous Transit in W/O Caller ID	[ ]		
Trunk Incoming Prompt Voice	(•) Default Greeting ( ) Recorded voice file ( ) Dial Tone		

- FXS Caller ID Generation: Select this option to enable the caller ID display function. When enabled, the caller's phone number will be displayed on your phone set when the call comes through. FSK is preferred in some countries.
- FXO Caller ID Detection: To detect the Caller ID delivered from PSTN to the FXO port. While enabled, the Caller ID detected on the FXO port will be send to the SIP Proxy Server on dialing out calls.
- Detection Level: If FXO can't detect Caller ID, try to adjust it until it can.

You have to enable **"**Wait for Caller ID before FXO / Trunk pick up" to ensure Caller ID is detected correctly**.**

- FSK Caller ID Type: Select FSK type. In most cases, Bellcore is preferred in North America and ETSI in Europe.
- Anonymous Caller ID (CLIR): When enabled, the caller's phone set will not display your number.

  If you register the VoIP gateway to a Proxy, you may be unable to make a call. This is due to the fact that the VoIP gateway doesn't send the number for authorisation.

- Anonymous Transit in W/O Caller ID: FXO won't detect caller ID, and the gateway will dial out with an anonymous caller identficiation. If the call needs caller ID to be identified for Proxy, Proxy will reject this call without caller id.
- Trunk Incoming Prompt Voice: Select the greeting (must use the IVR 132 function to record a voice file) when FXO receives an inbound call.

	Enable	Type	Hot Line	Hot Line No.	PSTN Busy-Out With FXS Pick-Up	Group Hunting	Grounding Compensation	Enable FAX
Line 1	☑	FXS	☐			☑		☑
Line 2	☑	FXS	☐			☑		☑
Line 3		FXO			9 [0,5-20s]		☐	
Line 4		FXO			9 [0,5-20s]		☐	

- Enable: To enable a line; if some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to this line.
- Hotline: When the user picks up the phone, Gateway automatically dials your assigned hotline number. When in hotline mode, other lines cannot be called.
- Hot Line No.: Enter the hot line number for an automatic dialing function.
- PSTN Busy-Out With FXS Pick-Up: It avoids calling in form FXO when FXS is getting DTMF. 0 is disabling.
- Group Hunting: Select group hunting when there is an incoming call, Gateway will automatically assign an unassigned call according to Hunting Priority. If Line 2 does not want to be set as an assigned line to receive any inbound call, the function can be disabled. Also, users can select the Up or Down key to adjust hunting priority (No setting is required for the FXO interface.
- Grounding Compensation: If the FXO can't work correctly- Noisy or Echo always in the back as you are talking.
- Enable FAX: Enable this line to detect is there FAX tone to transfer the Codec.

Wait for Caller ID before FXO / Trunk pick up	☑
Transit in Busy Tone Limit [0 - 60 s]	3
Ring Time Limit [10 - 600 s]	40
Enable End of Digit Tone	☐
Force Calling Thru PSTN Code	

- Wait for Caller ID before FXO / Trunk pick up: To detect caller ID from FXO port.
- Transit in Busy Tone Limit: The duration VoIP gateway plays a busy tone before FXO hook-on. To notify the caller from PSTN that this call is finished.
- Ring Time Limit(10 - 600secs)：The timeout to cancel a call when no one answers.
- Enable End of Digit Tone：The VoIP gateway will play a "Beep-Beep" tone to notify the call is in progress.
- Force Calling Thru PSTN code: Dial the code to get a PSTN line for dial out. For example: If you would like dial "23456789" through PSTN, just dial "*33 23456789"

SIP Settings

SIP Settings			
All Call through OutBound Proxy	☐		
OutBound Proxy IP / Domain		OutBound Proxy Port [1 - 65535]	5060

- All Call through OutBound Proxy：An outbound proxy server handles SIP call signaling as a standard SIP proxy server would. Furthermore, it receives and transmits phone conversation traffic (media) in between two talking gateways. This option tells the gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote gateway. This helps VoIP calls to pass through any NAT protected network without additional settings or techniques. Please make sure your VoIP service provider supports outbound proxy services before enable it.

E.164

International Call Prefix Digit	(optional)	
Country Code	(Other) (optional)	
Long Distance Call Prefix Digit	(optional)	
Area Code	(optional)	
E.164 Numbering	To Invite Proxy	☐
	Transform to Transit Out	☐
ENUM Header Exception	070	

- International Call Prefix Digit: Enter the International call prefix.
- Country Code: Users please select the desired country code.
- Long Distance Call prefix Digit: The long-distance prefix digit for making a long-distance call.
- Area Code: Please enter the area code.
- E.164 Numbering: To invite Proxy to follow the E.164 rule. It depends on the Proxy. **If you fail to make a call, please contact your ITSP**.

Enable Support of SIP Proxy Server / Soft Switch	☑		
Proxy Server IP / Domain	voip.sip.sip	Proxy Server Port [1 - 65535]	5060
Proxy Server Realm		TTL (Registration interval) [10 - 7200 s]	600
SIP Domain	voip.sip.sip	Use Domain to Register	☑
VoIP failure announcement	☑		

- Enable Support of SIP Proxy Server / Soft Switch: Enable the functions to inter-work with Proxy Server / Soft Switch.

- Proxy Server IP/Domain: Enter the Proxy Server IP address or URL (Uniform Resource Locator). You can set 3 redundant Proxy spread by semicolon.
  EX: 61.123.231.1; 12.34.56.78;proxy.sip.sip
- Proxy Server Port: Enter the Proxy Server **listen** port number. (The factory default value is 5060)
- Proxy Server Realm: Enter the correct registered Proxy Server Realm name to avoid registration failure. **If you fail to make a call, please contact your ITSP.**
- TTL: Enter the desired time interval at which the gateway will report to you Proxy Server
- SIP Domain/Use Domain to Register: Enter the correct SIP domain to avoid registration failure (it is not necessary to set with some Proxy Servers). If you enable "Uses Domain to Register" the VoIP gateway will register to Proxy with the domain name you filed. Else, the VoIP gateway will register to a Proxy with the IP it resolves. **If you fail to make a call, please contact your ITSP**.
- VoIP failure announcement: If a Gateway fails to register to the Proxy Server or the number that you dialed is not existed in Proxy Server, it will play a voice announcement when FXS is pick-up.

FXS Representative number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password
Gateway and SIP Prefix Number		10328536	☑		123	123

Assuming that your registered ID and password are individual, the settings as above:

- FXS Representative Number: To register all FXS ports as a hunting group.

Each line registers to Proxy independently:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password
Line 1	FXS	10328536 701 Auto	☑	☑	123	123
Line 2	FXS	10328536 702	☑	☑	123	123

- Invite with ID / Account: VoIP Gateway can invite to a VoIP Trunk Gateway w/o register to a Proxy. Please contact your ITSP

As there are various Proxy Server providers, our company has designed the Gateway to be compatible with them, and according to RFC standards. If any registration problem occurs, please consult your Proxy Server provider.

NOTE: When you register with a Proxy Server, dialing principles may vary with different Proxy Servers. Please consult your Proxy Server Provider.

Calling Features

Calling Features									
Line	Type	Unconditional Forward	Busy Forward	No Answer Forward	Forwarding Number	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID
FXS Representative Number		☐	☑	(N/A)	2252	(N/A)	(N/A)	(N/A)	
Line 1	FXS	☑	☐	☐ Forward after[10 - 60] 20 s	2253	☑	☑	☑	☑ 996688
Line 2	FXS	☐	☐	☑ Forward after[10 - 60] 20 s	2254	☑	☑	☑	☑ 996688

- Unconditional Forward: All incoming calls will be forwarded to the “Forwarding Number” automatically.
- Busy Forward: Forward the incoming call to “Forwarding Number” when the port is busy.
- No Answer Forward: Forward the incoming call to “Forwarding Number” after ring timeout expires without answer.
- Call Hold: Enable the call hold feature on the specific FXS port.
- Call Transfer: Enable the call transfer feature on the specific FXS port.
- Call Waiting: Enable the call waiting feature on the specific FXS port.
- Three-Way Calling / Service ID: Feature code of conference call defined on Nortel Soft Switch

Calling Feature Instructions:

- Call Hold: Ongoing call will be put on hold after FLASH button pressed on the phone set. The gateway will play a repeating music to the remote end.
- Call Transfer: Ongoing call will be put on hold after FLASH button pressed on local phone set (gateway plays a repeating music to the remote end). Meanwhile, local user can dial out to another number after dial tone observed. After the handset is back on the hook, the call on hold will then be transferred to the new call regardless of the status of the new cal. If wrong number is dialed for the new call, just press the FLASH button to get back the call on hold. In another case, if the local user doesn’t hang up the phone after new call sets up, press FLASH button to switch between the first call and the new call. Please be informed that PBX between phone sets and VoIP gateway must support FLASH features to make this function work correctly. If a phone set is connecting directly to the FXS port of the gateway and not functioning to FLASH, please adjust the settings in “Flash Detect Time” in category “Advanced Options”.
- Example of a Three-Way calling:
    1. Alex dials to Bob, Bob answers that call.
    2. Alex presses Flash and call to Coral (Bob is on hold), Coral answers that call.
    3. Alex dials *61 then presses Flash, thus conference call is created.

  Or

    1. Alex dials to Bob, Bob answers that call.
    2. Coral dials to Alex (Call Waiting), Alex presses Flash to pick the second call and talk to Coral.
    3. Alex dials *61 then presses Flash, thus conference call is created.

Advanced Options

Administrator's Name			
Administrator's Password		Confirm Password	
Web UI Login ID			
Web UI / IVR Password		Confirm Password	
Web UI auto logout [30 - 300 s]	60		

- There are two levels to enter Web. Administrator is able to change all settings. Web UI only changes some settings.
- Web UI auto log out: When logging in a web page, if a user does not act within the effective time range, the user will be disconnected from the web page to allow others to login.

Dial Wait Timeout[1 - 60s]	10	Inter Digits Timeout[1 - 60s]	4
Minimum DTMF ON Length[40 - 500ms]	80	Minimum DTMF OFF Length[40 - 500ms]	80
DTMF Detection Sensitivity	(less) ○ 1 ○ 2 ◉ 3 ○ 4 ○ 5 (more)		
Enable Out-of-Band DTMF ☐	◉ RFC 2833	Payload Type[96 - 127]	101
	○ SIP Info		
Use Second CPT for VoIP Call	☐		
Line Settings (Gain, Flash Time, Polarity Reversal)			

- Dial Wait Timeout: To set the waiting time for the user's first key pressing when dialing a number. The user will hear a busy tone if he/she does not press the first key within the effective time range.
- Inter Digits Timeout: Set the waiting time between each key pressing. The inputted numbers will be dialed after the timeout.
- Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones): used to set dial tone when a call is being diverted to other extension.
- DTMF Detection Sensitivity: used to adjust the sensitivity of the telephone keys.
- Enable Out-of-Band DTMF: To send DTMF keys (**0~9**, *****, **#**,) follow the RFC2833 rules or via SIP Info.
- Payload Type：Payload type of RFC2833.
- Uses Second CPT for VoIP Call: This function is usually applied when user select VoIP as the primary path for outgoing calls and PSTN as the backup. By enabling this function, the gateway will generate a different set of tones to inform user that VoIP is in service. While VoIP fails and fallback to PSTN, the user will hear PSTN tones instead of the second set CPT. (for CPT related settings please refer to Trunk Management -> CPT Settings)

Line Settings

Line Settings								
	Extension Number	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time	Enable Polarity Reversal	PSTN Answer Detection
Line 1	9901	FXS	0 All	0 All	5 All	0.6 All	☐	
Line 2	702	FXO	0	0	5	0.6	☐	Disable

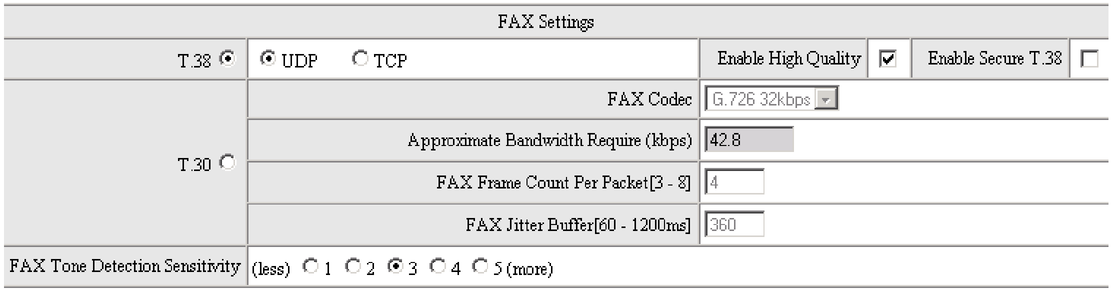
- Listening Volume: Adjust hearing volume.
- Speaking Volume: Adjust speaking volume.
- Tone Volume: Add a new option to make tone volume adjustable. This setting will be applied to all tones generated by the VoIP gateway including Dial Tone, Busy Tone, etc.
- Flash Time:
  FXS：To adjust the detecting period of flash signal from the phone set connected to the FXS port. For example, if pressing the HOLD key will disconnect a call, increase the "Flash Detect Time" should fix this issue.
  FXO：The time frame of FXO generate a FLASH signal.
- Enable Polarity Reversal:
  FXS：As the remote answer this call or hook on the FXS port will reverse the polarity.
  FXO：This option forces gateway to detect the reversal of polarity on FXO port as the primary signal to drop a call. Some telephone switches or PBX reverse the line polarity to inform remote end to drop an ongoing call. Please consult with the telephone service provider for availability of this feature.
- PSTN Answer Detection: This is used only by ITSP.

Voice

Voice				
Preferred Codec Type	G.723.1 6.3kbps			
Jitter Buffer[60 - 1200ms]	360			
Silence Suppression	☐	Echo Cancelling	☑	
Codec	G.711	G.723	G.726	G.729A
Packet Time (ms)	60	60	60	60
Approximate Bandwidth Require (kbps)	71.2	13.6	39.2	15.2

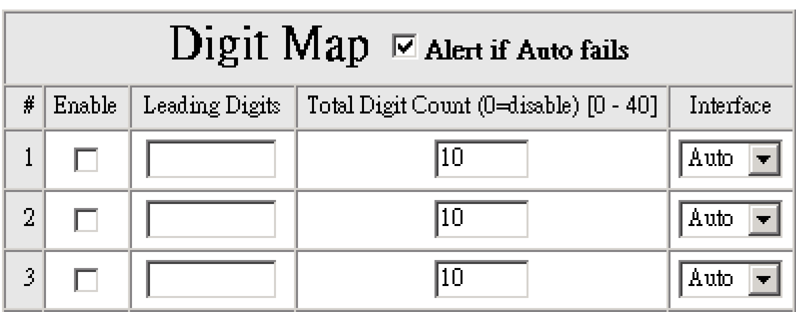
- Preferred Codec Type: Since different voice codec have different compression ratios, so the sound quality and occupied bandwidths are also different. It is recommended to use the default (G.723.1) because it occupies less bandwidth and will provide better sound quality.
- Jitter Buffer: Adjust the jitter to receive a packet. If the jitter range is too wide, it will delay voice transmission.
- Silence Suppression: If one side of a connection is not speaking, the system will stop sending voice data (package) to decrease bandwidth usage.
- Echo Canceling: Prevents poor telecommunication quality caused by echo interference.
- Packet Time: Define how long the VoIP Gateway sends a RTP packet-voice packet- to the other side. The smaller use more bandwidth, the larger you get more voice delay.
- Approximate Bandwidth Require: The bandwidth required varies with codec format and packet time.

Fax Settings



- T.38: The T.38 protocol is used for better and faster facsimile transmission. When this function is enabled, the following fax and voice parameter settings will be disabled, so it is recommended to enable this function to gain better fax quality. When this function is enabled, please select UDP, TCP, or AUTO. If selecting TCP and some routers cannot use the Fax function, please select UDP instead.
- Enable High Quality: The system sends the same FAX frame twice to get high quality of FAX. It requires more bandwidth.
- Enable Secure T.38: This use to send FAX for SP series Gateways on both sides.
- T.30: The system use T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the fax function will consume more network resources and will affect transmission quality.
- FAX Detect sensitivity：used to adjust the sensitivity of detect a phone call whether be FAX or not.

Digit Map



50 sets of leading digit entries to choose voice routing interface – Auto select, PSTN or VoIP.

- Alert if Auto fails: If Auto is selected; it will play a voice announcement before calling out. It remind user that this call is through PSTN.
- Enable: Enable detection of this entry.
- Leading Digits: The leading digits for VoIP Gateway to scan while user is dialing.
- Total Digit Count: Total number of digits that VoIP Gateway should accept. 0 disables this feature.
- Interface: The interface calls should go through if above conditions satisfied.

Local Phone Book

This system can set up 100 phone numbers into a phone book and provides an IP address query when calling to other Gateway(s). If no Phone books manager is set within a Gateway group, then all Gateway systems have to set up phone data for each VoIP Gateway to communicate with each other.

Local Phone Book 1 - 5				6 - 10
#	Gateway Name	Gateway Number	IP / Domain Name	Port
1				5060
2				5060

- Gateway Name: Please enter other Gateways code or an easy-to-remember name.
- Gateway Number: Enter the desired Gateway number.
- IP/Domain Name: Enter the IP address or URL (Uniform Resource Locator) of other Gateways.
- Port: Enter the Gateway listen port of other Gateways.

Speed Dial

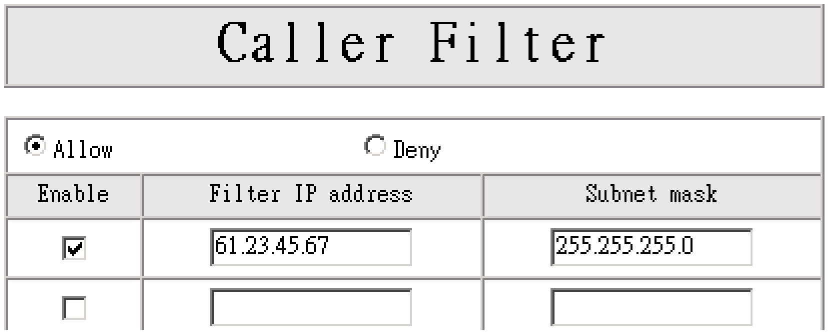
This system can set up 100 numbers for speed dialing. Setting methods are as follows:

Speed Dial 1 - 10	11 - 20	
#	Speed Dial Code("?" = single digit ; "%" = wildcard)	Number To Dial
Speed Dial 1	55	32568791
Speed Dial 2	3??	52137?
Speed Dial 3	00%	856%

Method 1: Single mapping. Fill a short code into the "Speed Dial Code" column, and enter the desired phone number into the "Number To Dial" column.
For example, pick up the handset and dial **55#** and the system will invite 32568791.

Method 2: Multi mapping. Fill prefix code into the " Speed Dial Code" column, and the format to transfer into the "Number To Dial" column.
For example, pick up the handset and dial **301#**, and the system will invite 521301.
If the user dial 00 1657987456321, the system will invite 856 1657987456321

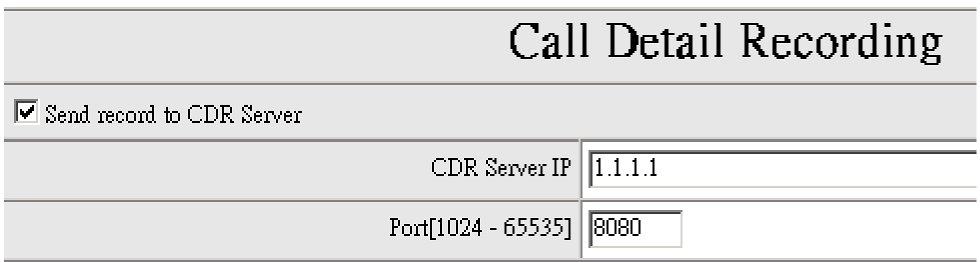
Caller Filter



To allow or deny SIP Invite from the list. The system always allows the invite that sent from the SIP Proxy it register to.

- Filter IP Address: Fill up with the start IP you would like to allow/deny.
- Subnet mask: Fill up with the subnet mask you would like to allow/deny.

CDR Settings



The user can set up a CDR Server to record call detail for every phone call.
The present CDR provides the call detail recording in a text file and imports the text file to prepare for an analysis report, if needed.

- Send record to CDR Server: Enable call detail recording function.
- CDR Server IP: Enter the IP address of the CDR server.
- Port: Enter the listen port of the CDR server.

Language

The system provides English, Traditional Chinese, and Simplified Chinese to display text on Web pages. Meanwhile, it will change the language for IVR (Interactive Voice Response).

CPT/Cadence Settings

CPT parameters table is below, and these parameters are for FXS playing Call Progress Tone.

# 1 Enable Detect Setting 1						Default
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	350	440	3000	0	0	0
Congestion Tone	480	620	250	250	0	0
Busy Tone	480	620	500	500	0	0
Ring-Back Tone	440	480	1000	2000	0	0

System Information

System Information

Port Status

No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	701		0		Disabled	
2	FXS	702		0		Disabled	
3	FXO	703		0		Disabled	
4	FXO	704		0		Disabled	

Server Registration Status

DDNS Registation	Disabled
Phone Book Manager Registation	Disabled
STUN Registation	Disabled
UPnP Negotiation	Disabled

RTP Packet Summary

Display info of the last finished call. It contains peer IP, peer port, packets sent, packet received and packet lost.

RTP Packet Monitor

Line 1	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
	The last packet's source IP			The last packet's source Port	0		
Line 2	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
	The last packet's source IP			The last packet's source Port	0		
Line 3	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	0

STUN Inquiry

STUN Inquiry	
NAT Type	Open
STUN Server	1.2.3.4
STUN Server Port(1024 - 65535)	3478

Ping Test

Ping Test	
IP address	192.168.8.59
Request Count[1 - 100]	4
PingTestPacketSize[56 - 5600bytes]	56

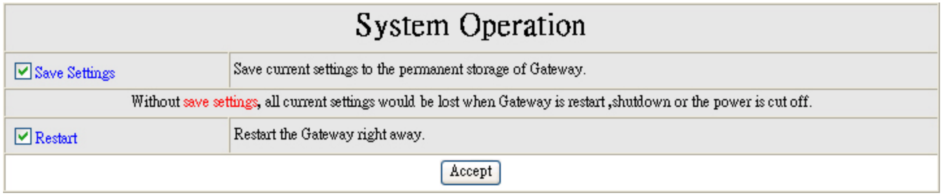
Use "ping" to identify if the remote peer is reachable. Fill in remote IP address and click "Test" will start the test.

Time Settings

Time Settings						
	Year	Month	Day	Hour	Minute	Second
IAD Time	2005	2	20	3	46	24
Time Zone	+ 0 : 00					
#	Time Server					
1	1.2.3.4					
2						
3						

Time Zone: Set the Time Zone where VoIP Gateway resides.
Time Server #1~#3: Set the Time Server where VoIP Gateway should sync up during start up. (NTP protocol)

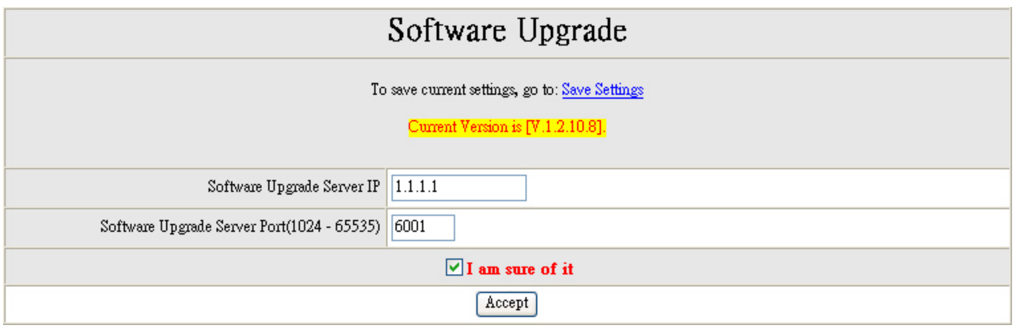
System Operations (Save Settings)



- Save Settings: Save settings after completing. The new settings will take effect after the system is restarted. Please select “Save Settings”.
- Restart: If it is necessary to restart the system, please select “Restart” and click the “Accept” button.

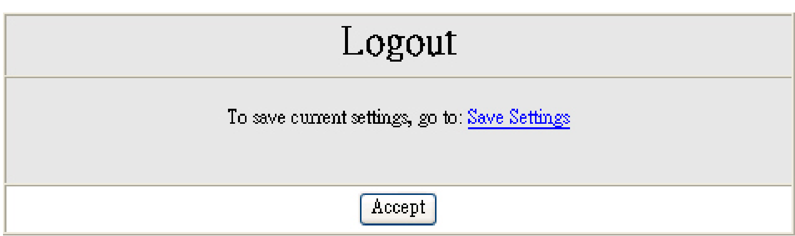
Software Upgrade

Gateway provides software upgrade function for a remote end.



- Software Upgrade Server IP address: Please enter the software server IP address.
- Software Upgrade Server Port: The default setting is 6001(Do not change this setting).
- Please select “ I am sure of it”, and click “Accept” button to update the software.

Logout



Gateway only allows one user to login at a time, so whenever a change is made, please save the settings, restart the system, or logout to avoid the situation where other users cannot login to change settings.

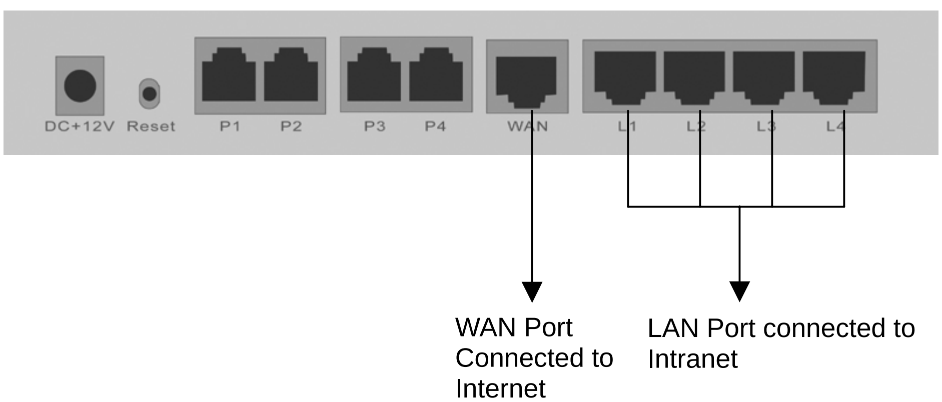
5. IP Sharing Functions

All Gateway series have a built-in IP sharing function. The settings and instructions at a PC end are described below:

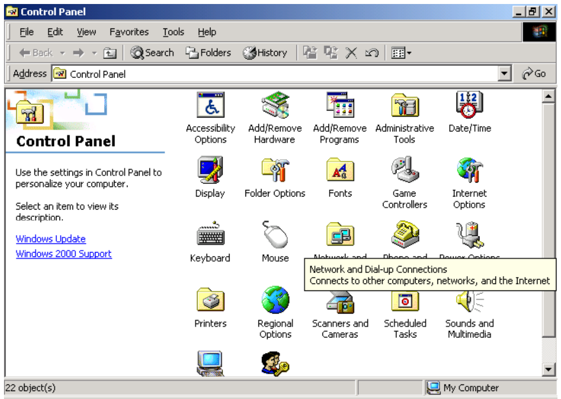
Current Intranet only supports static IP mode, and the settings at the PC end are as follow:
Available IP address Range : 192.168.8.1 – 192.168.8.253 (default address of Gateway is 192.168.8.254)
Subnet Mask : 255.255.255.0
Default Gateway : 192.168.8.254

The above values vary with different LAN Port Settings.
Assume Gateway's LAN settings are,
IP address : 192.168.3.1
Subnet Mask : 255.255.255.0

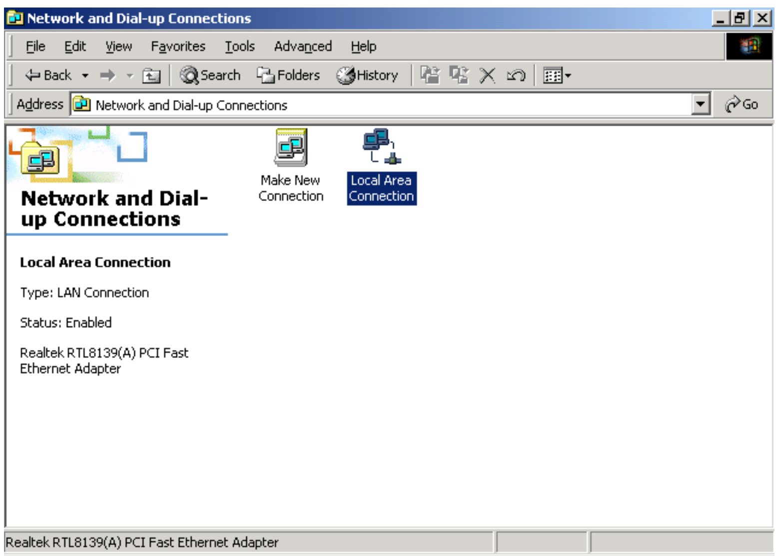
Then, the settings at PC end should be as follows:
Valid IP address range : 192.168.3.2 – 192.168.3.254
Subnet Mask : 255.255.255.0
Default Gateway : 192.168.3.1



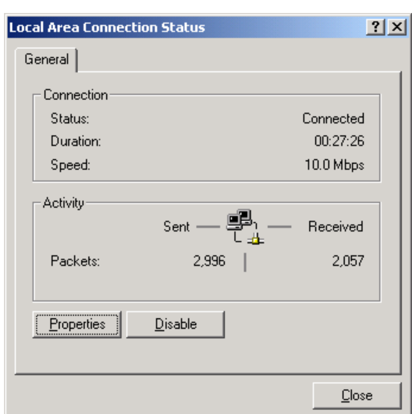
The IP settings on PC are as follows (using Windows 2000 for example)
Open Start->Settings->**Control Panel**
Open **Network and Dial-up Connection**



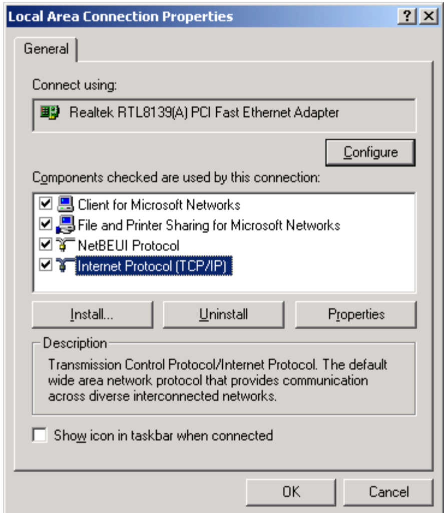
Open Local Area Connection



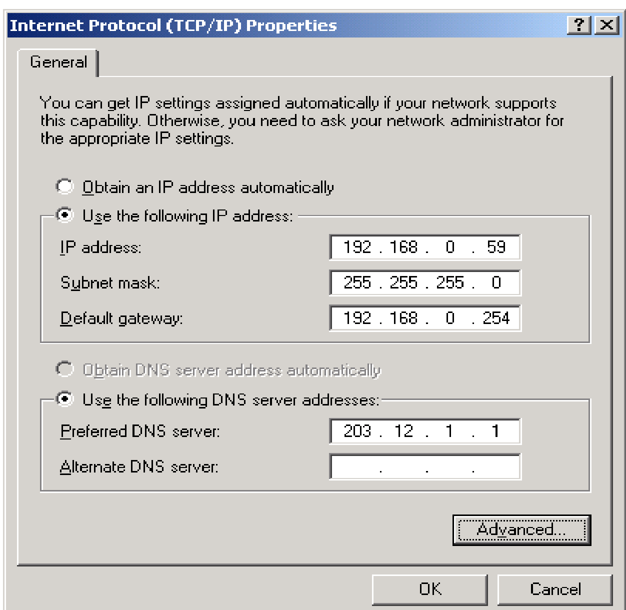
Click **Properties**



Select **TCP/IP**, and then click **Properties**.



Select "**Use the following IP Address "** and enter IP address, Subnet Mask, and Default Gateway. Please note that an IP address in the same domain cannot be reused. Then, enter the DNS server IP address (varies in different networks. consult your ISP's service for information). Click the "**OK"** button and after completing the settings, users can use both the VoIP and network services concurrently.



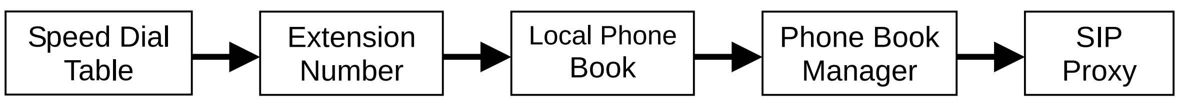
6. Coding Principle

Instruction

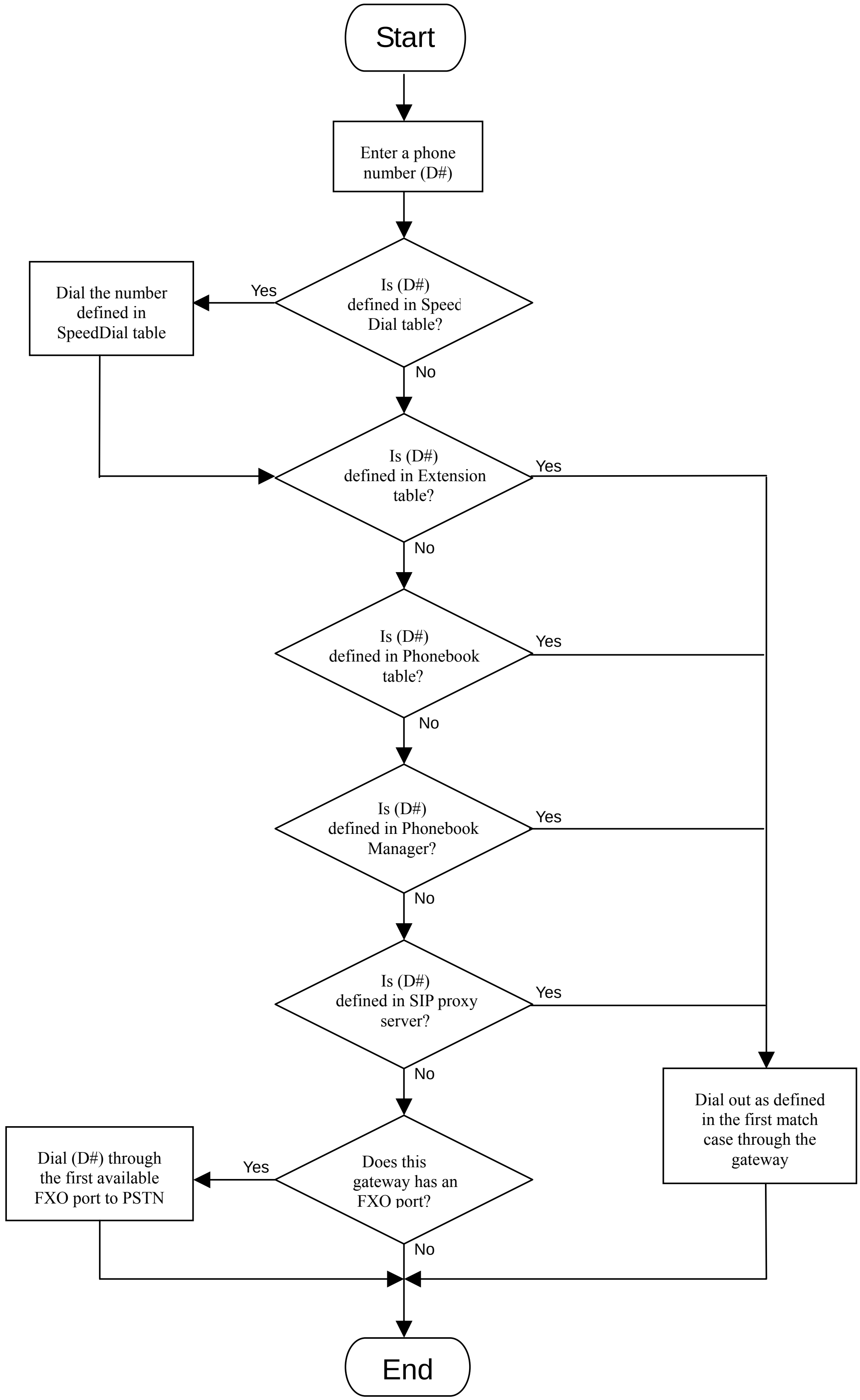
- After a phone number is entered, dial # to call out immediately or, wait until the "Inter DTMF Timeout" expires (defined in "Advanced Options", default=4 seconds).
- If the phone number fits the setting of Digit Map, the Gateway dials out the phone number through the assigned interface automatically.
- The phone number should have at least 2 digits (not including * and #).

Dialed Number Processing Flow

To maintain maximum flexibility, the number dialed will be looked up from several tables defined in the Gateway. Once no match can be found, it will look up again from the registered SIP Proxy Server. The look up flow is shown below:



A complete flow chart is on the next page.


Start
Enter a phone number (D#)
Is (D#) defined in Speed Dial table?
Yes
Dial the number defined in SpeedDial table
No
Is (D#) defined in Extension table?
Yes
No
Is (D#) defined in Phonebook table?
Yes
No
Is (D#) defined in Phonebook Manager?
Yes
No
Is (D#) defined in SIP proxy server?
Yes
No
Dial out as defined in the first match case through the gateway
Does this gateway has an FXO port?
Yes
Dial (D#) through the first available FXO port to PSTN
No
End

7. Advanced Feature

Static Route※

Static Route				
#	Route	Route Mask	Next Hop IP	Interface
1				
2				
3				
4				
5				

To build static routes within an internal network. These routes will not apply to the Internet.

- Route: Enter the IP of the specified network.
- Route Mask: Enter the subnet mask to be used for the specified network.
- Next Hop IP: Enter the IP address to the specified network.
- Inter Face: Select the inter face.

RIP(Routing Information Protocol) ※

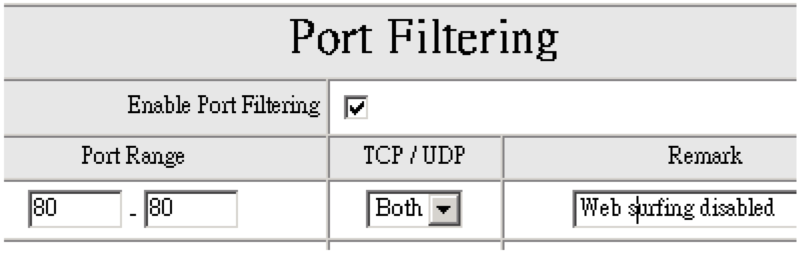
RIP	
Enable RIP	
Send Version	1&2
Receive Version	1&2
Enable Authentication	
Authentication Password	
Update Timer	30
Timeout Timer	180
Garbage Timer	120

To establish dynamic routes within an internal network. These routes will not apply to the Internet.

- Enable Authentication/Authentication Password: If the box is checked. All the boxes in this RIP group should be filled in the same password.

Port filtering

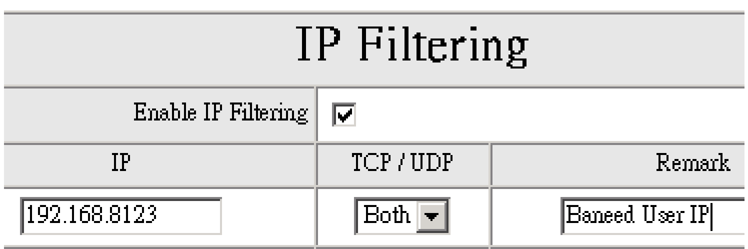
Port filtering enables you to control all data that can be transmitted in routers; **principles of filtering---When the port used at the source end is within the limited scope, it will be filtered without transmission.**



- Enable port filtering: whether to enable this function or not.
- Port Range: Set the range of port to be filtered, suppose it is 80 and when use protocol is Both or TCP, all computers will be unable to use the services of http (port 80)— will be unable to browse normal WebPages.
- TCP/UDP: Choose to either filter TCP or UDP, or choose to filter both.
- Remark: Remark field, you can write comments by yourself.

IP Filtering

IP Filtering is to limit internal users from accessing the Internet.



- IP: Input the IP address that you want to filter; the limited IP address will be unable to transmit the data to the Internet.
- TCP/UDP: Choose to either filter TCP or UDP, or choose to filter both.
- Remark: Remark filed, you can write comments by yourself.

MAC Filtering

MAC (Media Access Control) address filtering is to filter the transmission of data by network card physical address.

MAC Filtering	
Enable MAC Filtering	☑
MAC	Remark
000d12345678	Banned MAC address

MAC: input MAC that will limit accessing Internet PC.

Virtual Server

Enabling the users on Internet to access the WWW, FTP and other services under your NAT. When remote user are accessing Web or FTP servers through WAN end IP address, it will be routed to the server at the internal LAN end and be routed to the server at the internal LAN end as appropriate in accordance with the externally required services

Virtual Server				
Enable Virtual Server	☐			
WAN Port Range	TCP / UDP	LAN Host IP Address	Server Port Range	Remark
0 - 0	Both		0 - 0	
0 - 0	Both		0 - 0	

- WAN Port Range: Input the port on the WAN side.
- TCP/UDP: Select the communication protocols used by the server—TCP or UDP.
- LAN Host IP Address: Input IP address that provides various services servers.
- Server Port Range: Input the port used by the LAN host.

DMZ

DMZ	
Enable DMZ	☑
DMZ Host IP	192.168.8.123

Lets the server on the LAN to be directly exposed to the Internet for accessing data. Either this function or the virtual server can be selected for use.

URL Filter※

URL Filter			
Enable	URL string pattern to be blocked	Source IP Range	
☑	xxx.com.xx	192.168.8.1	- 192.168.8.253

URL filter is used to deny device from LAN accessing specific web sites. The system will block the URL that contains the string.

Special Applications※

Special Application					
Enable	Name	Incoming Type	Incoming Port Range	Trigger Type	Trigger Port Range
☐	MSN Gaming Zone	BOTH	2300-2400,28800-29000	TCP	47624 - 47624
☐	Quick Time	TCP	6970-6999	TCP	554 - 554
☐	ICU II	BOTH	2000-2038,2050-2051,2069,2085,3010	TCP	2019 - 2019

Provide multiple connections for special applications.

- Name: The name of the special application.
- Incoming Type: The protocol used to trigger the special application.
- Incoming Port range: Port range on the WAN side that will be used to access the application.
- Trigger Type: The protocol used to trigger the application.
- Trigger Port Range: Port range used to trigger the application.

DoS Prevention Settings※

DoS Prevention Settings		
Enable DoS Prevention	☑	
Enable DoS Prevention on LAN	☑	
Whole System Flood	☑ SYN	50 (Packets/Second) [50 - 500]
	☑ FIN	50 (Packets/Second) [50 - 500]
	☐ UDP	100 (Packets/Second)
	☑ ICMP	50 (Packets/Second) [50 - 500]
Per-Source IP Flood	☑ SYN	30 (Packets/Second) [30 - 300]
	☑ FIN	30 (Packets/Second) [30 - 300]
	☐ UDP	100 (Packets/Second)

- Enable DoS Prevention: To prevent DoS from WAN.
- Enable DoS Prevention on LAN: To prevent DoS from LAN.
- Packet/Second: If the same packet type is more than the setting in one second, then it will be attacked. And VoIP Gateway will block the IP if you checked “Enable Source IP Blocking”.
- Enable Source IP Blocking: Block the IP.
- Blocking Time: The time to block the IP.

Notice

※

These functions are only supported for the second hardware. The second hardware can support more functions, such as PPTP, LAN Interface Mode, LAN QoS, etc. There is an easiest way to make out the first and second hardware, please check Network Settings of User Interface. If there is LAN Interface Mode or PPTP in Network Settings, this model is the second hardware.

General

"IMPORTANT SAFETY INSTRUCTIONS - When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock and injury to persons, including the following:

-Do not use this product near water for example, near a bathtub, washbowl, and kitchen sink or laundry tub, in a wet basement or near a swimming pool.

-Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.

-Do not use the telephone to report a gas leak in the vicinity of the leak.

-Use only the power cord and batteries indicated in this manual. Do not dispose of batteries in a fire. They may explode. Check with local codes for possible special disposal instructions.

SAVE THESE INSTRUCTIONS"

Telephone line cord

"CAUTION: To reduce the risk of fire, use only No. 26 AWG or larger UL Listed or CSA Certified Telecommunication Line Cord"

END OF THIS DOCUMENT